AMENDMENTS TO THE CLAIMS

Claim 1 (Currently Amended): Digital data processing apparatus comprising a module (M2, M3, M_3 ")A method for zeroing a portion of a time domain impulse response of a filter in a speech for zeroing a portion of a time domain impulse response of a filter in a speech transmission apparatus which filter has a frequency domain transfer function Z(k), said method comprising:

implementing convolution with a function U on athe frequency domain data vector transfer function Z(k) where \underline{k} lies in the range 0 to N-1, which convolution corresponds to zeroing samples in the time domain of the inverse transform of Z(k), the apparatus being characterized in that wherein the function U has the form:

$$U(k) = \sin c \left(\frac{k - k_0}{2}\right) \cdot e^{-j\pi \left(\frac{\alpha(k - k_0)}{2}\right)} \cdot P(k)$$

where $\underline{\alpha}$ is a parameter, k_0 is a constant integer and P(k) is a weighting window that is symmetrical on about both sides of k_0 .

Claim 2 (Currently Amended): Apparatus The method according to the preceding claim 1, characterized in that wherein k_0 is equal to zero.

Claim 3 (Currently Amended): <u>The method Apparatus</u>-according to either one of the preceding claims 1, characterized in that it further comprises comprising a module (M₃')-receiving a frequency domain vector (H), the module being suitable for and inserting between two coefficients of the vector (H) on each occasion an additional coefficient so as to supply a frequency domain vector (H') of increased length.

Claim 4 (Currently Amended): <u>Apparatus The method</u> according to the <u>preceding claim 3</u> characterized in that wherein the additional coefficients are zeros.

Claim 5 (Currently Amended): <u>The method Apparatus</u> according to claim 3 or 4, characterized in that wherein for a frequency domain vector (H') of increased length

having indices of 0 to 2N-1, the inserted coefficients are the coefficients having odd indices.

Claim 6 (Currently Amended): Apparatus The method according to any one of claims $\underline{43}$ to 5, characterized in that wherein the module (M2, M3, M3") performing the convolution with U is placed realized downstream from the insertion module (M3'), and in that Z is the frequency domain vector of increased length (H').

Claim 7 (Currently Amended): Apparatus The method according to any preceding claim 4, characterized in that it includes a filter (H) upstream from the module (M2, M3, M3") for further including performing convolution with U upstream from inserting additional coefficients, and in that Z is the transfer function of this filter (H).

Claim 8 (Currently Amended): Apparatus-The method according to the preceding claim 7, characterized in that it includes means (H) for further including computing the coefficients of the filter (H) on the basis of a an input signal (X, S1) input to the apparatus.

Claim 9 (Cancelled)

Claim 10 (Currently Amended): Apparatus-The method according to claims 6, 7, and 9 in combination 1, characterized in that it further comprises comprising a first module (M2) applying convolution with a first function having the form:

$$U(k) = \sin c \left(\frac{k - k_0}{2}\right) \cdot e^{-j\pi \left(\frac{\alpha(k - k_0)}{2}\right)} \cdot P(k)$$

on a frequency transform (X) of an input signal (51) that is optionally augmented, and <u>applying</u> a second convolution module (M3) applying convolution with a second function having the form:

$$U(k) = \sin c \left(\frac{k - k_0}{2}\right) \cdot e^{-j\pi \left(\frac{\alpha(k - k_0)}{2}\right)} \cdot P(k)$$

on the frequency response (H) of a filter (H) that is optionally augmented, the output vectors (X', S3) from these two modules (M2, M3, M_3 ") having the same number of coefficients, and in that the apparatus has at its output a module (M4) suitable for method includes multiplying together the coefficients of these two output vectors (X', S3).

Claim 11 (Currently Amended): Apparatus The method according to any preceding of claims 4 to 6in combination with any one of claims 3 to 5, characterized in that wherein the module (M2, M3, M3") for performing of the convolution with U supplies an output vector (H', S3) having the same length as the augmented vector Z, by retaining in the output vector (H', S3) those coefficients of the vector Z which were present prior to insertion, and the other coefficients of the output vector (H', S3) being obtained by convolution of Z and U.

Claim 12 (Currently Amended): Apparatus-The method according to any preceding claim 1, characterized in that wherein the module (M2, M3, M3") for performing convolution with U outputs a vector B(0,...,N-1) which is such that for all \underline{k} , a coefficient in B of index \underline{k} is equal to a product of convolution between Z and U which is such that the coefficient of index k_0 in Z is multiplied in said convolution product with the coefficient of index k_0 of U for which the sinc function has an argument of 0.

Claim 13 (Currently Amended): Apparatus The method according to any preceding claim 1, characterized in that wherein the filter-function U takes non-zero values over a range of values of \underline{k} which is symmetrical about the value k_0 for which the modulus of U is at its maximum.

Claim 14 (Currently Amended): Apparatus-The method according to any preceding claim 1, characterized in that wherein the function U has an odd number of coefficients Lu, and in that U can be written:

$$U(k) = \sin c \left(-\frac{Lu-1}{4} + \frac{k}{2} \right) \cdot e^{-j\cot \left(-\frac{Lu-1}{4} + \frac{k}{2} \right)} \cdot P(k)$$

Claim 15 (Currently Amended): <u>Apparatus The method according to any preceding claim 1</u> in combination with claim 9 or 10, <u>characterized in that wherein</u> the transform is a discrete Fourier transform.

Claim 16 (Currently Amended): <u>Apparatus The method</u> according to <u>any</u> <u>preceding claim 1</u>, <u>characterized in that wherein</u> the weighting window is a Kaiser window having a coefficient of 1.5.

Claim 17 (Currently Amended): <u>Apparatus-The method</u> according to-any <u>preceding</u> claim 1, <u>characterized in thatwherein</u> it constitutes an echo <u>cancelling</u> <u>method</u> according to any <u>method</u> according to a method according to a method according to any <u>method</u> according to any <u>method</u> according to a method accordi

Claim 18 (Currently Amended): <u>Apparatus-The method</u> according to <u>any</u> <u>preceding-claim_1</u>, <u>characterized in thatwherein the methodit</u> constitutes a noise <u>reducing methodreducer</u>.

Claim 19 (Currently Amended): Apparatus The method according to any preceding claim 1, characterized in that wherein $\alpha = 1$.

Claim 20 (Currently Amended): Apparatus-The method according to any preceding claim 1, characterized in that wherein α =-1.

Claim 21 (Currently Amended): <u>An Aapparatus according to any preceding</u> claim, characterized in that it comprises comprising:

-a loudspeaker (100), a microphone (200), an echo canceller (420, 430, 440, 450), and a disturbance reducer (500), the echo canceller including an adaptive filter (470) and a subtracter module (300) delivering the error (Y') between a signal coming from the microphone (200) and a signal obtained by applying the adaptive filter (460) to a loudspeaker signal (100), the adaptive filter (460) adapting its coefficients as a function

of said error, and the apparatus including means (495) suitable for transforming the signal from the microphone into the frequency domain upstream from the subtracter module (300) in such a manner that the subtraction is performed in the frequency domain, wherein the apparatus includes means for implementing a method onto the adaptive filter, said method for zeroing a portion of a time domain impulse response of a filter in a speech for zeroing a portion of a time domain impulse response of a filter in a speech transmission apparatus which filter has a frequency domain transfer function Z(k), said method comprising:

implementing convolution with a function U on the frequency domain transfer function Z(k) where k lies in the range 0 to N-1, wherein the function U has the form:

$$U(k) = \sin c \left(\frac{k - k_0}{2}\right) \cdot e^{-j\pi \left(\frac{\alpha(k - k_0)}{2}\right)} \cdot P(k)$$

where α is a parameter, k_0 is a constant integer and P(k) is a weighting window that is symmetrical on both sides of k_0 .

Claim 22 (Currently Amended): <u>The Aapparatus</u> according to the preceding claim <u>21</u>, characterized in that it has further including means (430, 440) for transmitting the result of said frequency domain subtraction to the adaptive filter (470) of the echo canceller.

Claim 23 (Currently Amended): <u>The Aapparatus</u> according to claim 21, eharacterized in that wherein the disturbance reducer (500) is placed downstream from the subtracter module (300) and is applied in the frequency domain to the result of the subtraction.

Claim 24 (Currently Amended): <u>The Aapparatus according to the preceding</u> claim 23, characterized in that wherein the disturbance reducer (500) includes an adaptive filter (520) suitable for recalculating its coefficients as a function of a frequency domain input signal (Y') from the disturbance reducer (500).

Claim 25 (Currently Amended): <u>The Aapparatus according to the preceding</u> claim <u>24</u>, <u>characterized in that wherein</u> the disturbance reducer (500) is placed to receive the frequency domain signal (Y') output from the subtracter module (300) as said frequency domain input signal of the disturbance reducer.

Claim 26 (Currently Amended): <u>The Aapparatus</u> according to claim 24 or 25, characterized in that wherein the disturbance reducer (500) forms a loop receiving as input the frequency domain signal (Y) output from the subtracter (300), and applying at its output multiplication by the adapted coefficients of its adaptive filter on the frequency domain signal (Y') output by the subtracter (300).

Claim 27 (Currently Amended): <u>The Aapparatus</u> according to any one of claims 24 to 26, characterized in that wherein the same frequency domain signal (Y') is used as an error signal for adapting the adaptive filter (470) of the echo canceller and is multiplied by the coefficients of the adaptive filter (520) of the disturbance reducer (500).

Claim 28 (Currently Amended): <u>The Aapparatus according to any one of claims</u> claim 21-to 27, characterized in that whereiin no transform module is placed between the subtracter module (300) and the disturbance reducer (500).

Claim 29 (New): The method according to claim 1, wherein the filter is an adaptive filter of an echo canceller in an apparatus comprising a microphone and a loudspeaker.

Claim 30 (New): A method for zeroing a time domain speech transmission signal which signal has a frequency domain transform Z(k), said method comprising implementing convolution with a function U on the frequency domain transform Z(k) where k lies in the range 0 to N – 1, wherein the function U has the form:

$$U(k) = \sin c \left(\frac{k - k_0}{2}\right) \cdot e^{-j\pi \left(\frac{\alpha(k - k_0)}{2}\right)} \cdot P(k)$$

where α is a parameter, K_0 is a constant integer and P(k) is a weighting window that is symmetrical on both sides of K_0 .

Claim 31 (New): The method according to claim 30, further including a transform into the frequency domain of an input time domain signal (S1), upstream from performing convolution with U, and in that Z is said frequency transform (X), possibly associated with receiving a frequency domain vector (H), and inserting between two coefficients of the vector (H) on each occasion an additional coefficient so as to supply a frequency domain vector (H') of increased length.